User Manual
Model: IP Phone 202

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Recommendation:
Please read the entire manual before using your new IPPH202.

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1. WELCOME

IPPH 202 is an internet telephone set that features superb audio quality, rich functionalities, high level of integration, and compactness. By converting analog voice for transmission over the

internet, the IP Phone 202 allows users with broadband internet connections to make calls to and from anywhere in the world. The IPPH 202 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

2. WHAT IS IN THE PACKAGE

The IPPH 202 package contains:

1) One IPPH 202 VoIP phone
2) One IPPH 202 phone handset
3) One universal power supply
4) One Ethernet cable

3. PRODUCT OVERVIEW

3.1 Key Features

- Supports SIP 2.0
- Supports TCP/UDP/IP, RTP/RTCP, HTTP, ARP/RARP, ICMP, DNS, DHCP, NTP, TFTP protocols
- DHCP support for LAN or Cable modem
- PPPoE support for ADSL or Cable modem
- Supports IP address dynamic (DHCP) system
- Supports multi-method NAT traversal and anti-virus crossing
- Compliant with ITU-T standard and DTMF form and check
- Supports codecs G.723.1 (5.3K/6.3K), G.729A/B, iLBC, G.726, G.728, and wide-band G.722 sound coding and decoding
- Supports dynamic sound check VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Dynamic sound technology with sound quality as good as traditional telephones
- Advanced Digital Signal Processing (DSP) to ensure superb hi-fidelity audio quality
- Support standard voice features such as Caller ID Display or Block, Call Waiting, Hold, Transfer Forward, FLASH, in-band and out-of-band DTMF (RFC2833), Dial Plans, off-hook auto dial, configurable emergency dialing (e.g., 911), early dial, click-to-dial
- Advanced adaptive jitter buffer control, packet delay & loss concealment technology
- Support multi-net construction
- Auto–manual indication
- Built-in world clock auto time display
- Has single or dual network ports (10 BaseT)
- Support dial code regulation based on E.164
- Support TFTP remote upgrade
- English language IE setting
- Dialed, received number check
- Speed dial function
- Emulate telephone dialing
- Easy volume adjustment
- IP-to-IP and point-to-point communication
- HiFi low delay net sound communication
- Local IP number, local number check
- Instant plug and play
- Full duplex hands-free speakerphone superior frequency
- Call forwarding, call waiting, call hold, call transfer, redial, call log, volume control, voice mail with indicator

3.2 Hardware Specification

The table below lists the hardware specification of IPPH 202.

Model	IPPH 202
WAN interface	1 x RJ45 10 Base-T
PC interface	Available for IPPH 202B
Product Color	Almost Black or Grey
Universal Switching Power Supply	Input: 100-240VAC 50-60 Hz Output: +9VDC, 600mA, UL certified
Dimension	23cm (L) 19cm (W) 10cm (H)
All Package Weight	1.01kg
Operation Temperature	0 – 40° C, 32° –104° F
Storage Temperature	-30° –65° C, -22°– 149° F
Humidity	10 - 95%

4. INSTALLATION

1) Insert power adapter into back of the phone and connect it to a power outlet.
2) Connect internet cable from back of the phone (WAN Port) to broadband modem, router, hub, or switch.
3) For IPPH 202B, connect internet cable from back of the phone (PC Port) to a PC.

Warning: Please do not use a different power adapter. Using other power adapter may damage the IPPH 202 and will void the manufacturer warranty.

5 BASIC OPERATIONS

5.1 Get Familiar with Function Keys

Keypad Key	Function(s)
0 - 9, *(.), #	To input: numbers, *(.), #
+	Scroll to next menu/submenu selection, or lower the volume of the loudhailer
_	Scroll to previous menu/submenu selection, or rise the volume of the loudhailer
MENU	Enter menu mode
CALLED	Show the phone number that has been called out
CALLERS	Show the phone number that has been called in
MSG	Get the voice message or other messages
HOLD	Hold the present call for the moment

TRANSF	Transfer the present call to another number
CONF	Conference call for three sides
FLASH	Used to switch between two lines
MUTE/DEL	Delete the call history, voice message, etc.
REDIAL	Dial the last called number
SPEAKER	Enter the hand-free mode
Memory Key M1-M10	Under Development

5.2 Dialing or Making Calls

5.2.1 Two dialing modes:

Ordinary dialing mode (4 ways):

1. Pick the up handset or press the “SPEAKER” button.
2. Press the REDIAL button to dial the Dial the last called number.
3. View the CALLED history and choose to redial.
4. View the CALLER history and choose to redial.

IP address dialing mode:

1. Press “MENU” button to view the local IP address.
2. Dial the IP address and press “#” to start calling

5.2.2 Call Hold

While in conversation, pressing the “Hold” button will put the remote end on hold.
Pressing the “Hold” button again will release the previously Hold state and the bidirectional media will resume.

5.2.3 Call Flash

Assuming that call party A and party B are in conversation. Party A wants to speak to party C or if there is another incoming call:

1. Party A presses FLASH to get a dial tone.
2. Party A then dials party C’s number then #. Party A and party C now are in conversation.
3. By pressing FLASH button, Party A can choose to have conversation with B or C.

5.2.4 Call Transfer

5.2.4.1 Blind Transfer

The user can transfer an active call to a third party without announcement.
Assuming that call party A and party B are in conversation. Party A wants to transfer party B to party C:

1. Party A presses TRANSF to get a dial tone.
2. Party A then dials party C’s number then #.
3. Party A then press REDIAL. A quick confirmation tone if the transfer has been successful. Party B and party C now are in conversation.

4. Party A can hang up or make another call.

5.2.4.2 Attended Transfer

The user can transfer an active call to a third party without announcement.
Assuming that call party A and party B are in conversation. Party A wants to transfer party B to party C:

1. Party A presses Flash to get a dial tone.
2. Party A then dials party C's number then #.
3. If the call is answered, press "flash" to complete the transfer operation
4. if the call is not answered, pressing "flash" button to resume the original call.

5.2.5 3-Way Conference Call Assuming that call party A called party B, and now they are in conversation. Party A wants to let party C to attend the conversation:
1. Party A presses CONF to get a dial tone.
2. Then press party C's number then # (or wait for 4 seconds). Party A and party C now are in conversation.
3. Party A presses CONF again, now the 3 parties are all in conversation.
Note: The server needs to support this function for this feature or the phones to work.

5.3 Receiving Calls

When phone rings from incoming calls and to receive the incoming calls, pick up the receiver or press the "SPEAKER" button. After picking up the call, to switch from hand free mode to receiver mode, just pick up the receiver and begin talking; the speakerphone will be deactivated. To switch from receiver mode to hand free mode, press down the "SPEAKER" button while placing the receiver back to resting location. Now the speakerphone is activated and you may continue speaking without receiver.

5.4 Call Features

Following table shows the new call features of IPPH 202.

Key	Call Features
*70	Do Not Disturb. When someone call you, "do not disturb" busy tone will be heard. To use this feature, pick up the handset and dial "*70". It will be cancelled when you hang up.
*72	Unconditional Call Forward. To use this feature, dial "*72" and get the dial tone. Then dial the forward number and "#" for a dial tone, and then hang up.
*73	Cancel Unconditional Call Forward. To cancel "Unconditional Call Forward", dial "*73" and get the dial tone, then hang up.
*90	Busy Call Forward. To use this feature, dial "*90" and get the dial tone. Then dial the forward number and "#" for a dial tone, and then hang up.
*91	Cancel Busy Call Forward. To cancel "Busy Call Forward", dial "*91" and get the dial tone, then hang up.
*92	Delayed Call Forward. To use this feature, dial "*92" and get the dial tone. Dial the forward number and "#" for a dial tone and then hang up.
*93	Cancel Delayed Call Forward. To cancel this feature, dial "*93", get the dial tone, and then hang up.
Flash/Hook	When in conversation, this action will switch to the new incoming call if there is a call waiting indication. When in conversation without an incoming call, this action will switch to a new channel to make a new call.

6 PROGRAMMING IPPH 202

6.1 Default Manager (Administrator) Password: admin

6.2 Major Sets of Configuration Data

There are 8 major sets of data related to the programming of the IP Phone 202: IP Type, IP Address, Subnet Mask, Router, DNS Server1, DNS Server2, Account, and Password.

The IP Phone 202 needs to be assigned an IP address in order for it to access the internet and make telephone calls. There are 2 kinds of IP address types (IP Type):

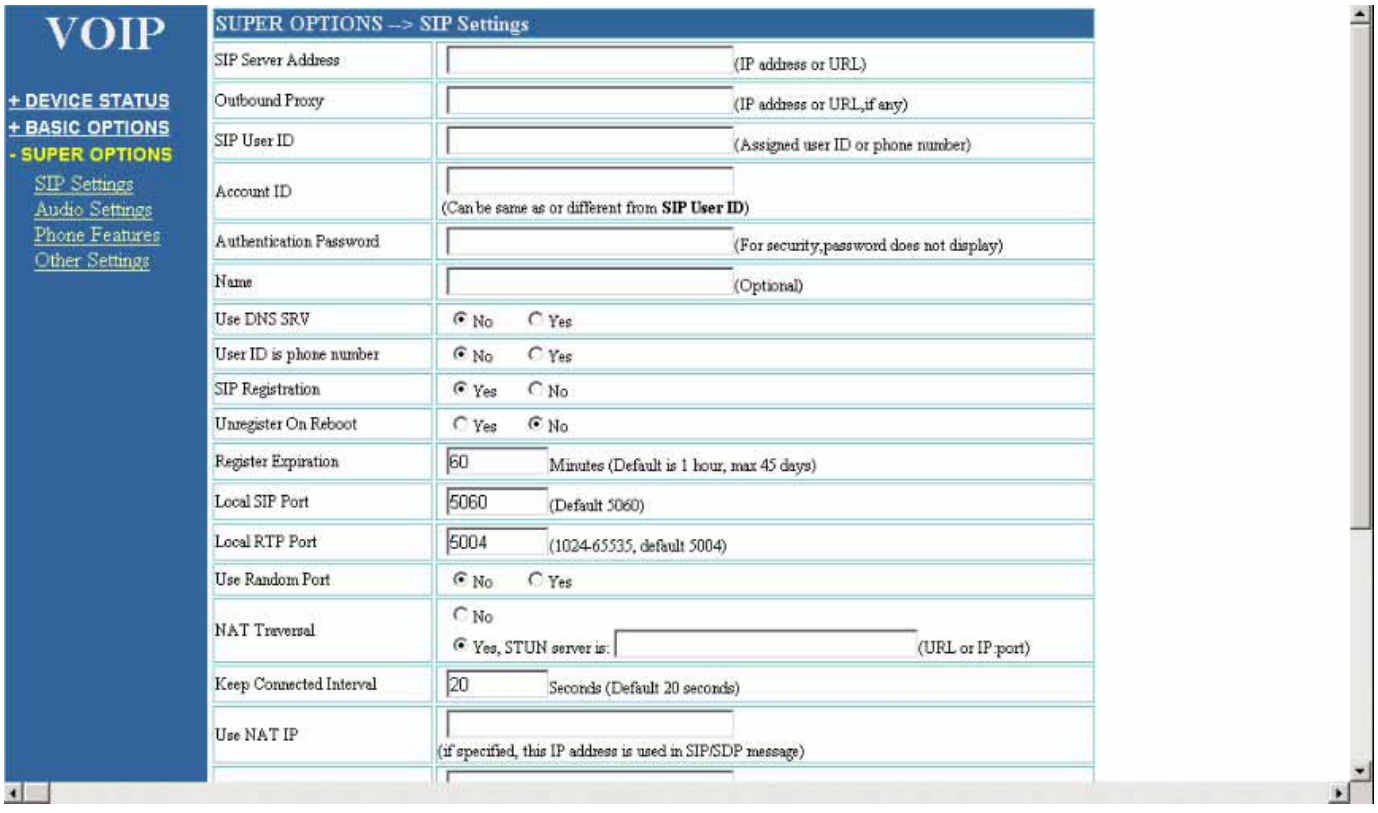
1) IP Type 0 (Static): IP address is manually entered by you. There are 5 fields to be filled:
   IP Address: 192.168.0.160 (default)
   Subnet Mask: 0.0.0.0
   Default Router: 0.0.0.0
   DNS Server1 (DNS using): 0.0.0.0
   DNS Server2 (DNS spare): 0.0.0.0

2) IP Type 1 (DHCP): IP address is dynamically assigned by router, if any, at the location, thereby allowing it to access the internet and make calls. Plug the IP Phone 202 into the router using the LAN port connection. The IP address is assigned to the IP Phone 202 automatically.

3) User Account: Assign your own user name
4) User Password: Assign your own password

6.3 Programming Using Web Browser

IPPH 202 has embedded Web server and HTML pages that allow users to configure the IPPH 202 through an easy-to-use Web browser interface such as Microsoft's Internet Explorer or Netscape browser. Below is a screen shot of the IPPH 202 configuration page:



6.3.1 Determine your VoIP device's IP address.

On the IPPH202 (while handset is up)
push the "MENU" key to obtain the phone's IP address, which will show on the LCD screen. If the IP address is all "0"s, then your VoIP device does not have internet connection. Please go back to step 1 and check your installation before proceeding.

6.3.2 User Programming and Configuration

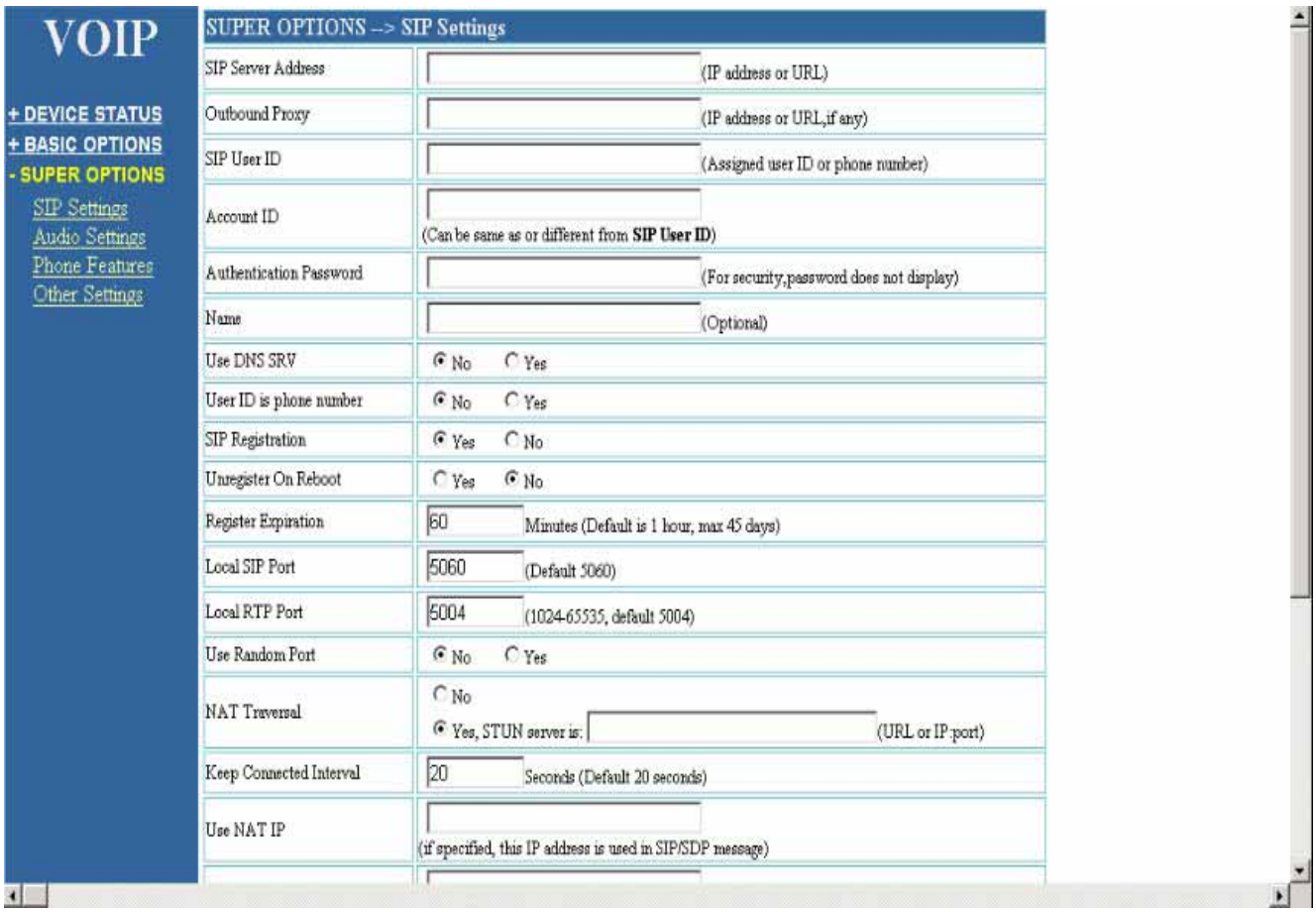
From your web browser, the IPPH 202 will show the following login screen:

Password: admin

6.3.3 Configuration Options and Explanations

After a correct password is entered in the login screen, the embedded web server inside the IPPH 202 will show the configuration page, which is explained in details below:

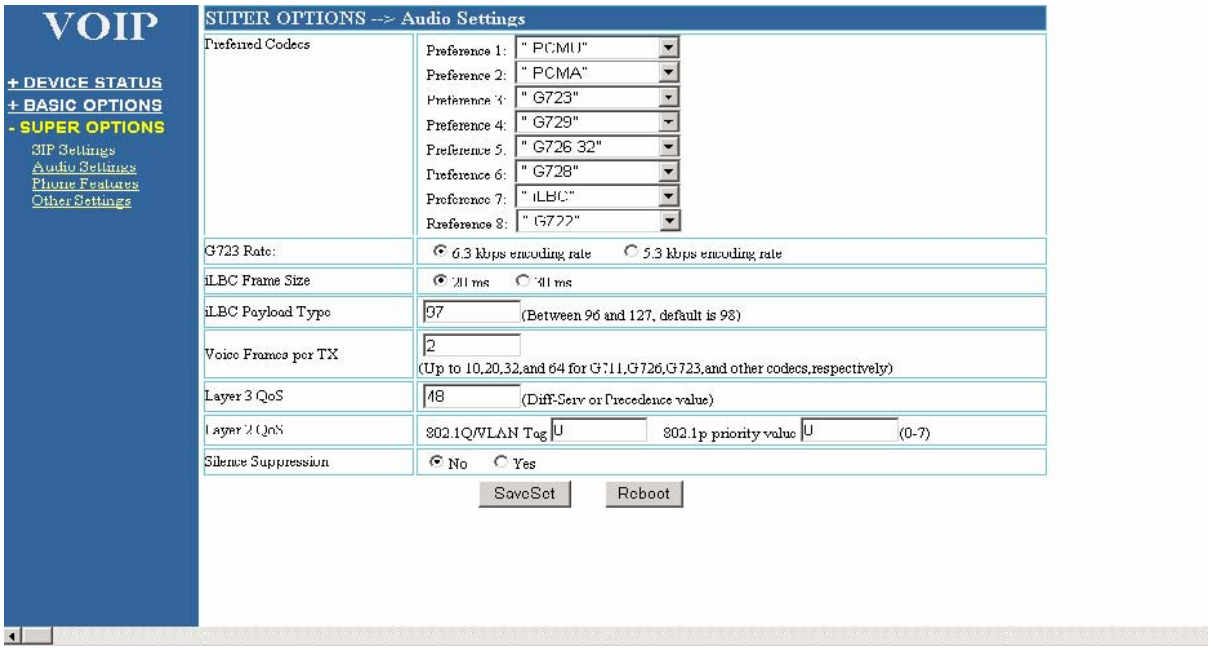
SIP Settings:



SUPER USER OPTIONS PAGE	
SIP Settings	
Options	**Meaning**
SIP Server Address	Gatekeeper's / SIP Server's URL or IP address
Outbound Proxy	SIP Outbound Proxy Server's URL or IP address
SIP User ID	SIP service subscriber's User ID
Account ID	SIP service subscriber's Authenticate ID. Can be identical to, or different from, SIP User ID
Authentication Password	SIP service subscriber's account password(for security, the password does not display)
Name	SIP service subscriber's name which will be used for Caller ID display
Use DNS SRV:	Default is No. If set to Yes, then the client will use DNS SRV for server lookup
User ID is Phone Number	If the IPPH 202 has an assigned PSTN telephone number, then this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request message to SIP server
SIP Registration	This parameter controls whether the IPPH 202 needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is "No." If set to "Yes", then the SIP user will be unregistered on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the IPPH 202 refreshes its registration with the specified gatekeeper/SIP server. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Local SIP Port	This parameter defines the local SIP port the IPPH 202 listens and transmits signals. The default value is **5060**.

Local RTP port	This parameter defines the local RTP-RTCP port pair the IPPH 202 will listen and transmit signals. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value is **5004.**
Use Random Port	When set to "Yes", this parameter will force random generation of both the local SIP and RTP ports. **This is usually necessary when multiple IPPH 202 are behind the same NAT.**
NAT Traversal	This parameter defines whether the IPPH 202 NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a **STUN server address** is also specified, then the IPPH 202 will automatically follow the STUN client specifications. Under this mode, the embedded STUN client inside the IPPH 202 will attempt to detect if there is a (and type of) firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port- Restricted Cone, the IPPH 202 will attempt to use its mapped public IP address and port in all of its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the IPPH 202 will periodically (every 20 seconds) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
Keep Connected Interval	This parameter specifies how often the IPPH 202 sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open.
Use NAT IP:	NAT IP address used in SIP/SDP message. Default is blank.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Send DTMF	This parameter controls how DTMF events are transmitted. There are 3 ways: In-Audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833
Send Anonymous	If this parameter is set to "Yes", the "From" header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Send Flash Event	This parameter allows users to control whether to send a SIP NOTIFY message indicating that the phone's Flash key has been pushed, or just to switch to the voice channel when users press the Flash key.

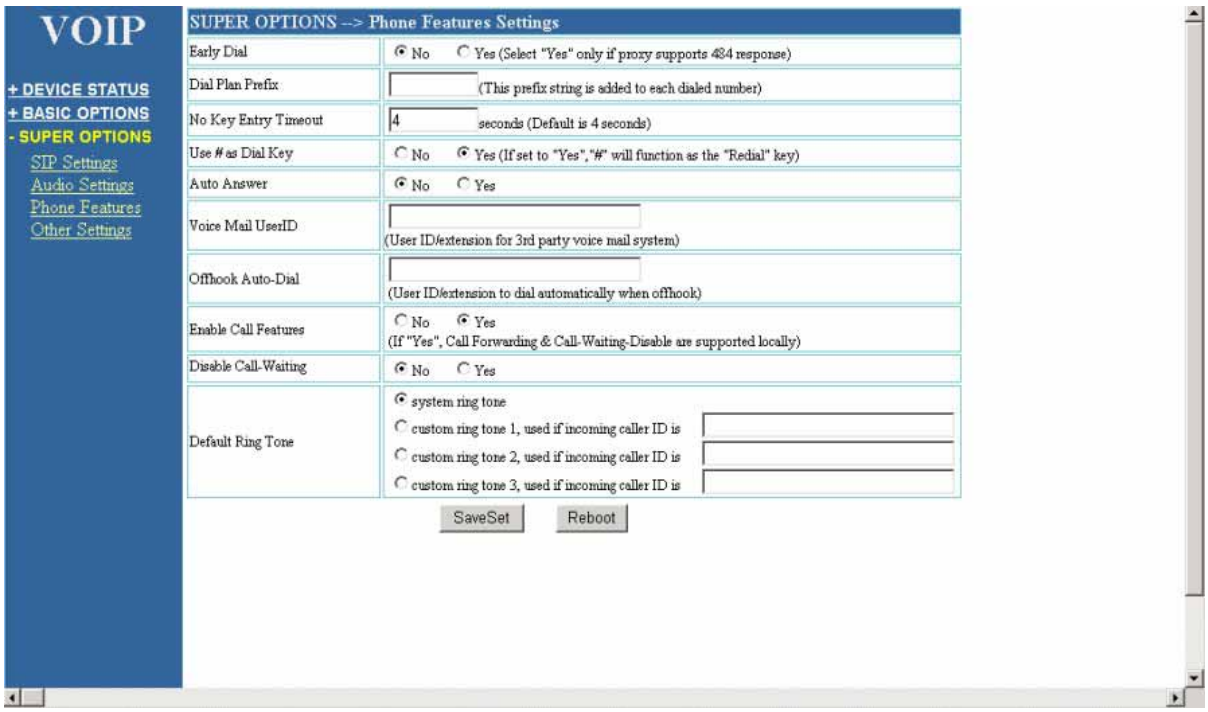
Audio Settings:



Audio Settings	
Options	**Meaning**

Preferred Codec	The IPPH 202 supports up to 7 different Codec types including G.711 G.723.1, G.726, G.728, G.729A/B, iLBC. Users can configure Codecs in a preference list that will be included with the same preference order in SDP message. The first Codec in this list can be entered by choosing the appropriate option in “Choice 1”. Similarly, the last Codec in this list can be entered by choosing the appropriate option in “Choice 7”.
G723 Rate:	This defines the encoding rate for G723 codec. By default, 6.3kbps rate is chosen.
iLBC Frame Size:	This defines the size of the iLBC codec frame. The default setting is 20ms.
iLBC Payload Type:	This defines payload time for iLBC. Default value is 97.
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first codec in the above codec Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. For example, if the first codec is configured as G723 and the “Voice Frames per TX” is set to be 2, then the “ptime” value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first codec chosen is G729 or G711 or G726, then the “ptime” value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the IPPH 202 will use and save the maximum allowed value for the corresponding first codec choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames, respectively.
Layer 3 QoS	This field defines the layer 3 QoS parameter, which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48.
Layer 2 QoS	This contains the value used for layer 2 VLAN tag. Default setting is 0.
Silence Suppression	This controls the silence suppression/VAD feature of G723 and G729. If set to “Yes”, when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking to make the connection feel more "natural". If set to “No”, this feature is disabled.

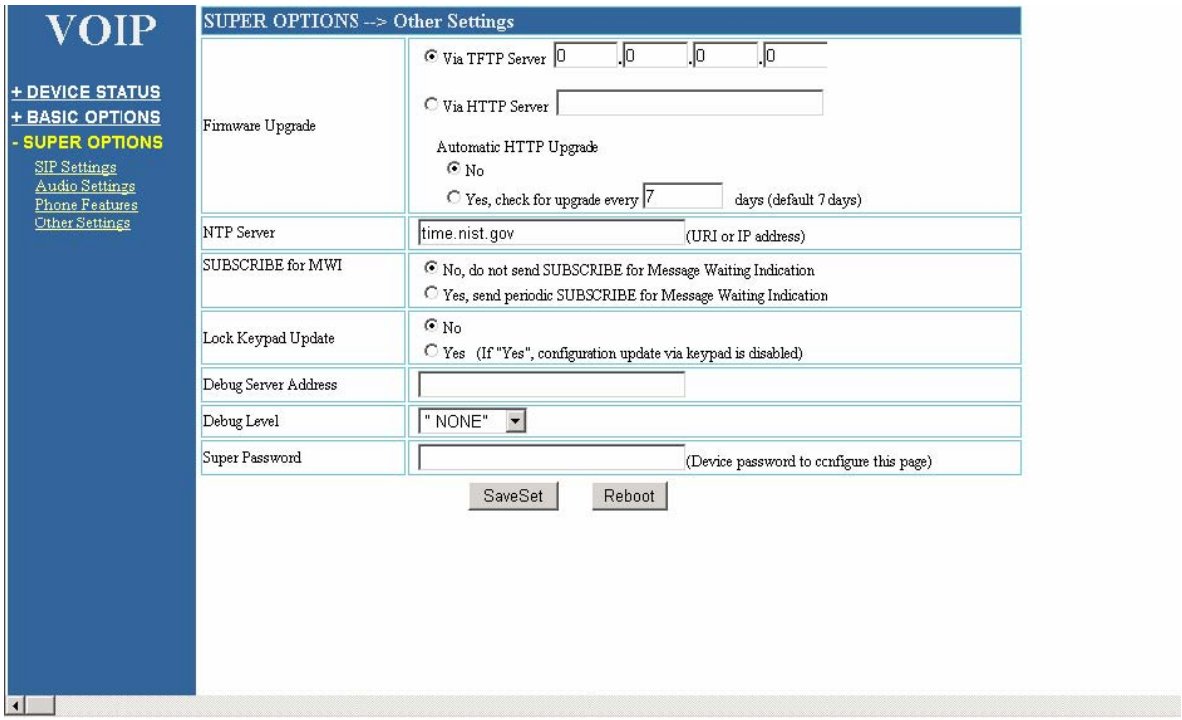
Phone Features Settings:



Phone Features Settings

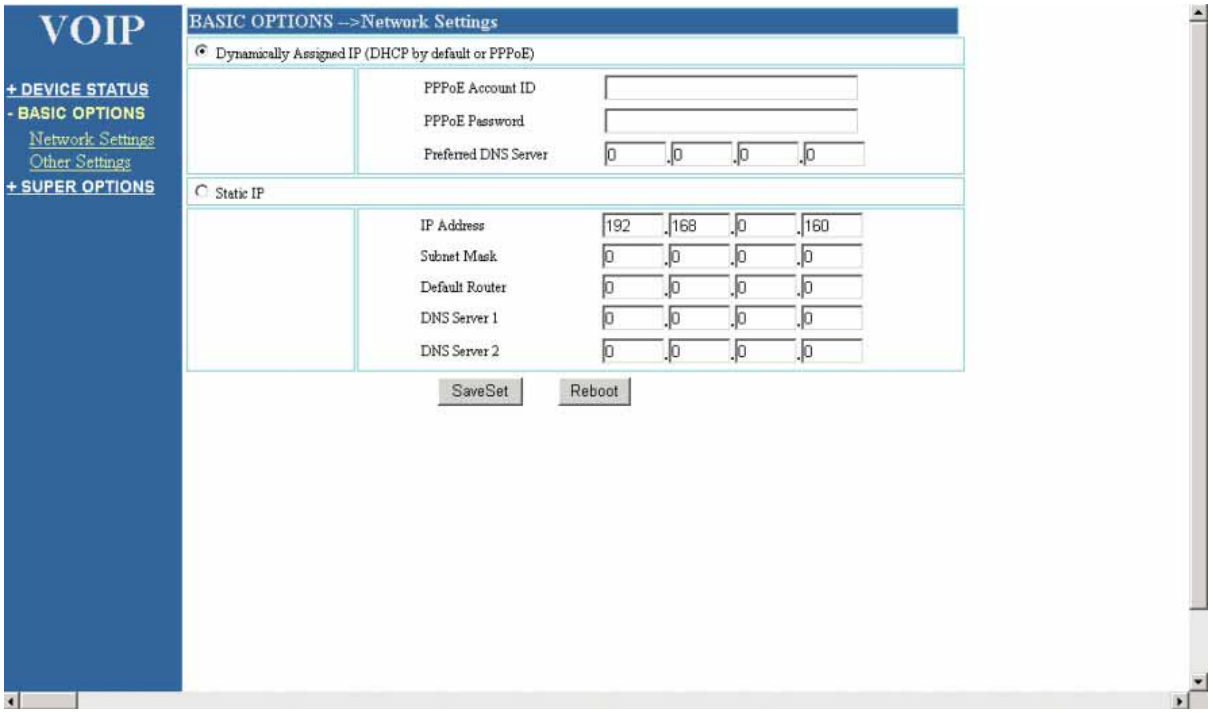
Options	Meaning
Early Dial	Default is “No”. Use only if proxy supports 484 response
Dial Plan Prefix	Sets the prefix added to each dialed number
No Key Entry Timeout	Default is 4 seconds. This is the time lapse after entering a phone number for the IPPH 202 to begin completing your call.
Use # as Dial Key	This parameter allows users to configure the “#” keyto be used as the “Send / Enter” (or “Dial”) key. If set to “Yes”, pressing this key will immediately trigger the sending of dialed numbers entered thus far. In this case, this key is essentially equivalent to the “(Re)Dial” key. If set to “No”, this “#”key will then be included as part of the dial string to be sent out.
Auto Answer Voice Mail	Default is NO. When set to yes the phone will automatically pick up the call and turn the speaker on.
Voice Mail UserID	This parameter defines the User ID (or extension number) of a 3rd party voice mail system where the user may have an account. By defining this Voice Mail extension, when the user presses the “Message” button on the phone, an INVITE message will be sent to that extension to allow the user to retrieve messages.
Off-hook Auto-Dial	This parameter allows users to configure a User ID or extension number to be automatically dialed upon off-hook. Please note that only the user part of a SIP address needs to be entered here. The IPPH 202 will automatically append the “@” and the host portion of the corresponding SIP address.
Enable Call Features	Default is “No”. If set to “Yes”, Call Forwarding & Do-Not-Disturb are supported locally
Disable Call Waiting	Default is “No”.
Default Ring Tone	The user can setup up one number on each of the ring tones 1, 2 and 3. When a call is received from one of these numbers, the respective ring tone will be generated.

Other Settings:



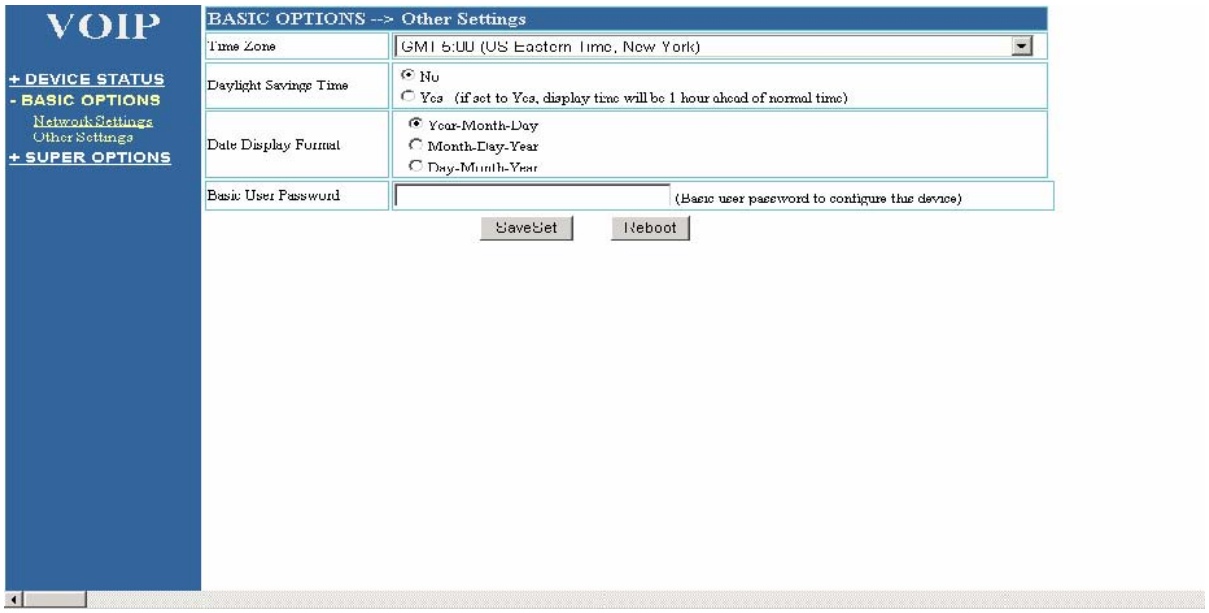
Other Settings	
Options	**Meaning**
Firmware Upgrade via TFTP server	The IP of TFTP server needs to be configured. If it is non-zero or not blank, the IP phone will attempt to retrieve new configuration file or new code image from the specified tftp server at boot time. It will make up to 3 attempts before timeout and then it will start the boot process using the existing code image in the Flash memory. If a tftp server is configured and a new code image is retrieved, the new downloaded image will be verified and then saved into the Flash memory. Be careful when the TFTP is in progress (the phone back light on LCD screen will be up). Do NOT interrupt the process (especially the power supply) otherwise might cause the device dead. Be patient as in some network this process will take up to 15 minutes.
Firmware Upgrade via HTTP server	The URL for the HTTP server used for firmware upgrade and configuration via HTTP. For example, http://upgrade.myVoIPservice.com:6688/IPPH202 Here ":6688" is the specific TCP port that the HTTP server is listening to, it can be omitted if using default port 80. Note: Auto Upgrade has to be set to "Yes" to enable HTTP upgrade.
Auto Upgrade	Choose Yes to enable HTTP upgrade and provisioning. In "Check for new firmware every ___ days" field, enter the number of days to check the HTTP server for firmware upgrade or configuration.
NTP Server	This parameter defines the URI or IP address of the NTP server which the IP phone will use to synchronize the current date/time.
SUBSCRIBE for MWI	Default is "No". When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.
Lock Keypad Update	If this parameter is set to "Yes", the configuration update via keypad is disabled.
Debug Server Address	This parameter defines the URI or IP address of the Debug server.
Debug Level	Default is "None". Choose the debug level you want when using this function.
Super Password	Default is "**Admin**". This contains the password to access the Advanced Web Configuration page. This field is case sensitive.

Network Settings



Network Settings	
Options	**Meaning**
Dynamically Assigned IP	If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory) and the IP phone will acquire its IP address from the first DHCP server it discovers on the LAN it attaches to. PPPoE account settings can be configured here too if the user is using DSL/ADSL connection. Users can specify DNS server manually by entering DNS server's IP address.
Static IP	If Static IP mode is selected, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary and mandatory), DNS Server 2 (secondary and optional) fields will need to be configured. These fields are reset to zero by default.

Other settings:



Other Settings	
Options	**Meaning**
Time Zone	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.
Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to Yes, then the displayed time will be 1 hour ahead of normal time.
Date Display Format	This parameter controls the date display format.
Basic User Password	This contains the password to access the Basic Web Configuration page. This field is case sensitive.

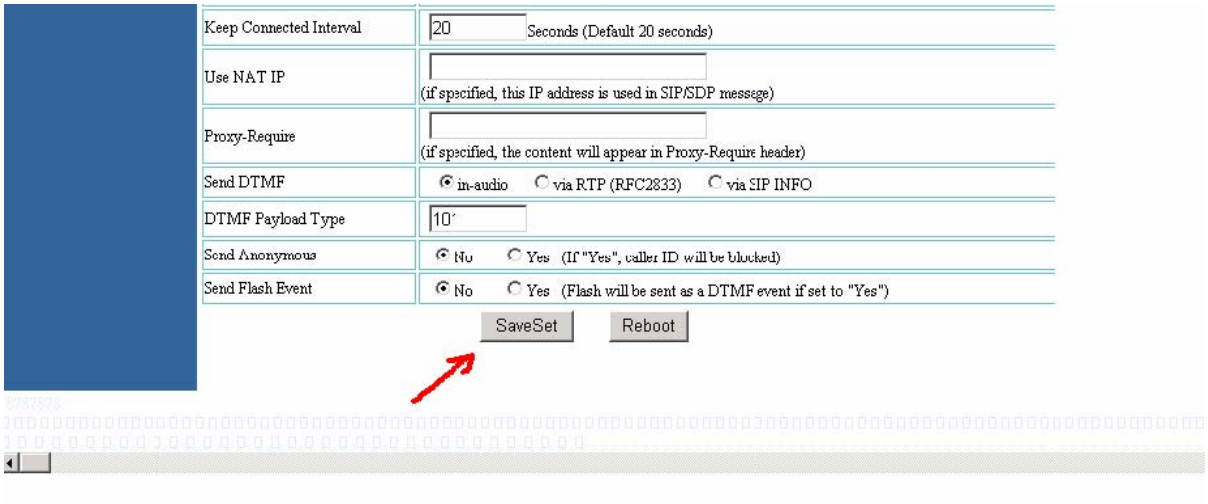
Device Status

VOIP
- DEVICE STATUS
+ BASIC OPTIONS
+ SUPER OPTIONS
Status
MAC Address 00.09.45.70.88.86
WAN IP Address 192.168.1.126
Software Version 2.0.6.5 (2005-09-19)
System Up Time 0 day(s) 0 hour(s) 2 minute(s)
Registered No
PPPoE Link Up disabled
NAT detected NAT type is open Internet
NAT Mapped IP 0.0.0.0
NAT Mapped Port 0
Total Inbound Calls 0
Total Outbound Calls 0
Total Missed Calls 0
Total Call Time 0 (Minutes)
Total SIP Message Sent 0
Total SIP Message Received 0
Total RTP Packet Sent 0
Total RTP Packet Received 0
Total RTP Packet Loss 0
Reboot

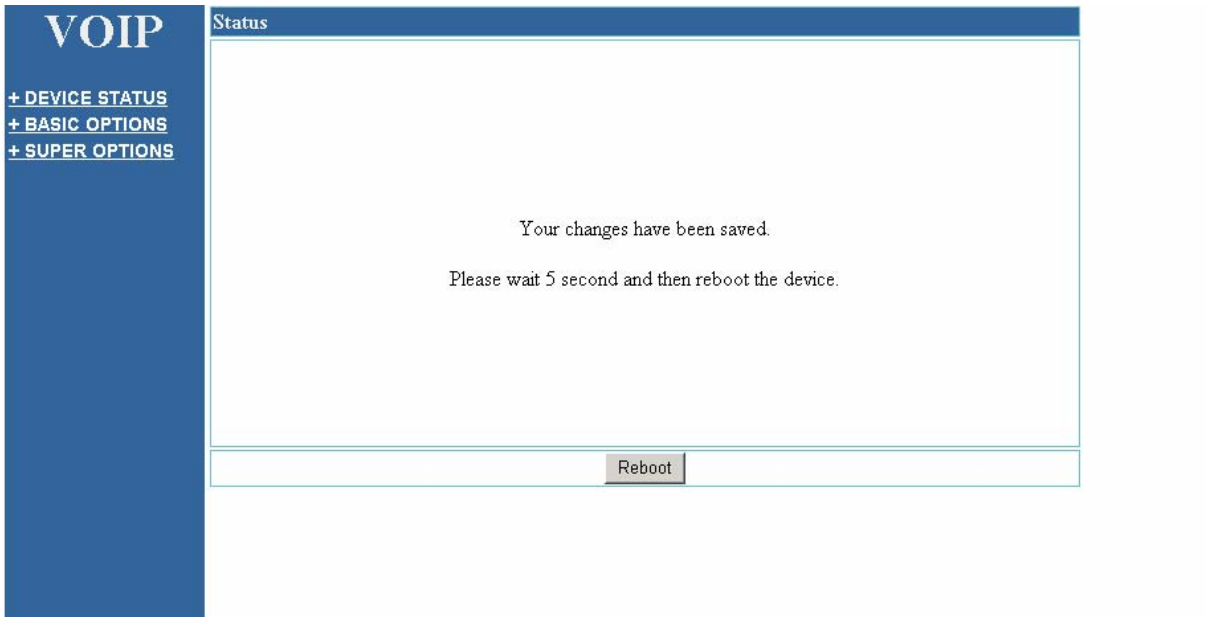
Device Status	
fields	**Meaning**
MAC Address	The device ID, in HEX format. This is very important ID for ISP troubleshooting.
WAN IP Address	This field shows WAN port IP address.
Software Version	This field shows the software version.
System Up Time	This shows system up time since last reboot.
Registered	This shows whether the unit is registered to service provider's server.
PPPoE Link Up	This shows whether the PPPoE is up if connected to DSL modem.
NAT	This shows what kind of NAT the BudgeTone is behind if it is not in oper interne.
NAT Mapped IP	WAN side public IP if connected to LAN of a SOHO NAT router.
Other Status	Please refer to the page displayed.

6.3.4 Saving the Configuration Changes

Once a change is made, users should click on the "SaveSet" button in the Configuration page.



The IPPH 202 will then display the following screen to confirm that the changes have been saved. *Please wait 5 seconds and then reboot the device.*



6.3.5 Rebooting the IPPH 202

You can reboot the IPPH 202 by clicking on the "Reboot" button after each update to the configuration page. Alternatively, you can reboot by unplugging the power supply of the IPPH 202 and then powering it on again. If your IPPH 202 ever becomes "stuck" or un-responsive, you can unplug the power supply to reboot it. Frequent rebooting by unplugging the power supply is not recommended and should not be necessary.

6.4 Programming Using Keypad

Configure with keypad

Menu Item	Menu function(s)

1	Show“ 1 dhcp on” Press“ 1 dhcp off”as the current option Press MENU and enter the menu mode Press “+”or“-”to choose Press MENU to save or exit Restart to be effective
2	Show “ 2 IP Addr” Press MENU to show the current IP address If DHCP is OFF enter new IP address Press“+”or“-” to exit Press MENU to save or exit Restart to be effective
3	Show“ 3 SubNet” Press MENU to show subnet address If DHCP is OFF enter new subnet address Press“+”or“-”to exit Press MENU to save or exit Restart to be effective
4	Show“ 4 routEr ” Press MENU to show router/gateway address If DHCP is OFF, enter router/gateway address Press“+”or“-”to exit Press MENU to save or exit Restart to be effective
5	Show“ 5 dnS ” Press MENU to show DNS address If DHCP is OFF, enter new DNS address Press“+”or“-”to exit Press MENU to save or exit Restart to be effective
6	Show“ 6 tFtP ” Press MENU to show tFtP address Enter new tFtP address Press“+”or“-”to exit Press MENU to save or exit
7	Show “[7] G-711u 2” Press MENU to show current codec Press ‘+’ or ‘-’ to view all codec options 1 “ - G-711A 2” 2 “ - G-722 2” 3 “ - G-723 1” 4 “ - G-726 2” 5 “ - G-728 8” 6 “ - G-729 2” 7 “ - iLBC 1” Press 1 to 9 to choose the frame number Press MENU to exit. No need to restart

8	Show“ 8 SIP SP-1” Press MENU to show SIP server/ service provider Press“+”or“-”to view all usable SIP server (1-9) Press MENU to save and exit SIP server must be configured through Web browser
9	Show “[9] codE rEL” Press Menu to display the code releases Press ‘+’ or ‘-’ to view 1 “b 2004-06-12” – date: boot code 2 “ 1. 0. 0.18” – version: boot code 3 “P 2004-06-21” – date: phone code 4 “ 1. 0. 5. 3” – version: phone code 5 “c 2004-05-06” – date: vocoder 6 “ 1. 0. 0. 6” – version: vocoder 7 “h 2004-06-17” – date: web server 8 “ 1. 0. 0. 36” – version: web server 9 “1r 2004-05-12” – date: 1st ring tone 10 “ 1. 0. 0. 0” – version: ring tone 11 “2r 2004-05-12” – date: 2nd ring tone 12 “ 1. 0. 0. 0” – version: ring tone 13 “3r 0000-00-00” – date: 3rd ring tone 14 “ 0. 0. 0. 0” – version: ring tone (all show 0 means not usable or not supported) Press Menu to exit
10	Show“ 10 Phy Addr” Press MENU to show MAC address Press“+”or“-”to exit
11	Show “[11] ring 0” Press MENU to hear current ring tone Press ‘+’ or ‘-’ to choose ring tones 4 ring tones available at present ring0 default ring1,ring2.ring3 are not usable or not supported Press MENU to choose and save Be effective directly. No need to restart.
	Show “ -- rESEt --”, ATTENTION: A> Enter MAC address and press MENU IPPH 202 will restart and get back to all default setting. All former configuration will be lost. B> Press MENU and the phone will restart.
Others	When in the mode: • Press‘+’ or ‘-’ and show“ ring [4] ”, press ‘+’ or ‘-’ to adjust the ring volume form 0 to 7 • Press “SPEAKER” , or pick up handset, then press ‘+’ or ‘-’ to adjust voice volume.

7. TEST YOUR SETUP

If connection is successful, you can begin using your VoIP phone. Your VoIP service provider will provide you with calling instructions. Due to varying dialing schemes of each provider, we do not

have specific dialing to recommend.

8. TECHNICAL SUPPORT CONTACTS

e-mail: office@alinea.bg